AMENDMENTS TO THE CLAIMS

The following listing of claims will replace all prior versions and listings of claims in the application.

LISTING OF CLAIMS

1. (Original) A system to compensate for the effects of packet delay on a voice over internet protocol (VoIP) system, comprising:

a buffer for receiving speech packets in the VoIP system;

a playback device for adjusting the playback speed of the received speech packets; and

a buffer manager for detecting packet jitter in the buffer and for sending commands to the playback device to adjust playback speed based on the detection.

2. (Original) The system of claim 1, wherein the buffer is a queue for handling incoming speech packets, the buffer performing jitter buffering and packet sequencing on the received speech packets.

3. (Original) The system of claim 1, wherein the buffer manager controls the playback device to decrease the playback speed when the buffer manager detects packet jitter that delays arrival of a speech packet.

4. (Original) The system of claim 3, wherein the buffer manager controls the playback device to increase the playback speed when the delayed packet arrives at the buffer.

5. (Original) The system of claim 3, wherein packet jitter is a variation in packet delay that causes packets to arrive out of sequence at an end-point in the system.

6. (Original) The system of claim 5, wherein an end-point is a client in the system.

7. (Original) The system of claim 1, wherein the buffer manager checks length of the buffer and instructs the playback device to increase playback speed until the length of the buffer returns to a nominal length, when the buffer manager determines that length of the buffer exceeds a specified length.

8. (Original) The system of claim 5, wherein the buffer manager detects packet jitter, the buffer manager measures a distance between an earliest detected out of sequence packet and a beginning of the buffer, and the buffer manager controls the playback device to decrease the playback speed when the distance is less than a predetermined distance.

9. (Original) The system of claim 1, wherein the buffer manager detects packet jitter, the buffer manager measures a distance between an earliest point of detected packet jitter in the buffer and a reference point in the buffer, and the buffer manager controls the playback device to decrease the playback speed when the distance is less than a predetermined distance.

10. (Original) The system of claim 1, wherein the buffer manager includes silence compression means that uses silence periods that are received between speech packets in the buffer to restore the length of the buffer to a nominal length.

11. (Original) The system of claim 10, wherein the buffer manager compresses the silence periods to return playback speed to a nominal speed.

12. (Original) The system of claim 1, wherein the buffer manager controls the playback device to adjust speed by an amount that is dependent on an expected or observed packet jitter.

13. (Original) The system of claim 12, wherein packet jitter is a variation in packet delay that causes packets to arrive out of sequence at an end-point in the system.

14. (Original) The system of claim 13, wherein an end-point is a client in the system.

15. (Currently Amended) A method of compensating for the effects of packet delay on a voice over internet protocol (VoIP) system, comprising the steps of:

detecting an amount of jitter in the arrival of speech packets; ~~and~~

adjusting a playback speed for the speech packets based on the detection; and

sending commands related to adjusting the playback speed to a playback device.

16. (Original) The method of claim 15, further comprising performing jitter buffering and packet sequencing on the speech packets prior to performing the detecting step.

17. (Original) The method of claim 15, wherein jitter is represented by an out of sequence packet, and wherein playback speed is decreased upon detection of an out of sequence packet, and increased upon arrival of the out of sequence packet.

18. (Original) The method of claim 17, wherein a length of a buffer storing the speech packets is checked against a specified length, and playback speed is increased until the length of the buffer returns to a nominal length, if the checked buffer length is less than the specified length.

19. (Original) The method of claim 17, wherein out of sequence packets are detected, a distance between an earliest detected out of sequence packet and a beginning of the buffer is measured, and playback speed is decreased when the measured distance is less than a predetermined distance.

20. (Original) The method of claim 15, wherein packet jitter is detected, a distance between an earliest point where packet is detected and a reference point is measured, and playback speed is decreased when the measured distance is less than a predetermined distance.

21. (Original) The method of claim 15, further comprising the step of restoring the length of a buffer storing the incoming speech packets to a nominal length, at a nominal playback speed, instead of at a higher playback speed, thereby compressing any silence intervals.

22. (Original) The method of claim 21, wherein the compressing step is performed when silence suppression is enabled in the VoIP system.